Grandstream Networks, Inc.

Analog IP Gateway GXW400X
4 or 8 FXS Ports

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GXW400X USER MANUAL

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http://www.grandstream.com/products/gxw_series/gxw400x/documents/gxw400x_gui.zip

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GNU GPL INFORMATION

GXW400X firmware contains third-party software licensed under the GNU General Public License (GPL). Grandstream uses software under the specific terms of the GPL. Please see the GNU General Public License (GPL) for the exact terms and conditions of the license.

Grandstream GNU GPL related source code can be downloaded from Grandstream web site from: **http://www.grandstream.com/support/faq/gnu_gpl** .

CHANGE LOG

This section documents significant changes from previous versions of GXW400X user manuals. Only major new features or major document updates are listed here. Minor updates for corrections or editing are not documented here.

CHANGES FROM 1.0.13.3 USER MANUAL

- Added feature **[Hold Target Before ReferHold Target Before ReferHold Target Before ReferHold Target Before ReferHold Target Before ReferHold Target Before ReferHold Target Before ReferHold Target Before Refer]** in profile settings, which allows user to hold or not hold the phone call before refer.
- Added feature **[Crypto Life Time]** in profile settings, which allows user to enable or disable Crypto life time when using SRTP.
- Added feature **[Play busy/reorder tone before Loop Current Disconnect]** in profile settings, which allows user to configure if it will play busy/reorder tone before loop current disconnect upon call fail.

CHANGES FROM 1.0.12.4 USER MANUAL

- Added option **[SIP Timer D]** to configure RFC 3261 timer D in Advanced Settings.
- Changed option name from **[Allow DHCP Option 66 to override server]** to **[Allow DHCP Option 66 or 160 to override server]**. Now option160 will be accepted by unit along with option 66 when enabled.
- Added feature **[Download Device XML Configuration]** in advanced settings, which allows user to download device configuration to local directory in xml format.
- Added feature **[Upload firmware]** in advanced settings, which allows user to upload firmware file from local directory.
- Added feature **[Upload configuration]** in advanced settings, which allows user to upload configuration file from local directory.

CHANGES FROM 1.0.12.1 USER MANUAL

- Added the options to enable/disable **[Always send HTTP Basic Authentication Information]**
- Added the options to restrict the m field sent in SDP **[Disable Multiple m line in SDPDisable Multiple m line in SDP]**

CHANGES FROM 1.0.11.3 USER MANUAL

- Added the options to enable/disable [**Do Not Escape '#' as %23 in SIP URI**]
- Added network whist/black list function on WAN port [**White list for WAN side**] [**Black list for WAN side**]

CHANGES FROM 1.0.10.9 USER MANUAL

- Added the options to enable/disable [**Use P-Preferred-Identity Header**] and [**Use Privacy Header**]
- Added the option to enable/disable [**Add Auth Header On Initial REGISTER**]

CHANGES FROM 1.0.7.6 USER MANUAL

- Added option to enable/disable SIP NOTIFY Authentication. [**Disable SIP NOTIFY Authentication**]
- Added option [**Use Configured IP**] in DNS mode. Added configurable parameter [**Primary IP**][**Backup IP1**][**Backup IP2**]
- Added option to set [**Reregister before Expiration**]
- Added option [**Use Request Routing ID in SIP INVITE Header**] to allow user to replace From and Contact headers for outgoing calls by [**Request URI Routing ID**]
- Added field [**Request URI Routing ID**] to allow device to route the calls to individual fxs ports based on the Request URI user ID in the incoming INVITE.

CHANGES FROM 1.0.6.8 USER MANUAL

- Added the option to enable/disable hook flash. [**Enable Hook Flash**]
- Added two CPE SSL configuration [**CPE SSL Certificate**][**CPE SSL Private Key**]
- Added a configuration parameter to set the [**Connection Request Port**]
- Removed DHCP Domain from Web UI
- Removed Enable Ring-transfer from Web UI
- Added a configuration parameter to set the [**Offhook Auto-Dial Delay**]
- Changed the SSL Web UI description to [**SIP TLS Certificate**][**SIP TLS Private Key**] and [**SIP TLS Private Key Password**]

WELCOME

Thank you for purchasing the Grandstream GXW400X Analog FXS IP Gateway. The GXW400X offers an easy to manage, easy to configure IP communications solution for any business with virtual and/or branch locations. The GXW400X supports popular voice codecs and is designed for full SIP compatibility and interoperability with third party SIP providers, thus enabling you to fully leverage the benefits of VoIP technology, integrate a traditional phone system into a VoIP network, and efficiently manage communication costs.

This manual will help you learn how to operate and manage your GXW FXS Analog IP Gateway and make the best use of its many upgraded features including simple and quick installation, multi-party conferencing, and direct IP-IP Calling. This IP Analog Gateway is very easy to manage and scalable, specifically designed to be an easy to use and affordable VoIP solution for the small – medium business or enterprise.

GATEWAY GXW400X OVERVIEW

The new GXW400X series has a compact and quiet design (no fans) and offers superb audio quality, rich feature functionality, strong security protection, and good manageability. It is auto-configurable, remotely manageable and scalable.

The GXW400X features 4 or 8 port FXS interface for analog telephones, dual 10M/100Mbps network ports with integrated router, PSTN life line in case of power failure, and an RS232 serial for administration. In addition, it supports the option of 3 SIP Server profiles, caller ID for various countries/regions, T.38 fax, flexible dialing plans, security protection (SIPS/TLS), comprehensive voice codec including G.711 (a/u-law), G.723.1, G.726(16/24/32/48 bit rates), G.729A/B/E and iLBC.

SAFETY COMPLIANCE

The GXW400X is compliant with various safety standards including FCC/CE. Its power adapter is compliant with UL standard. **Warning:** use only the power adapter included in the GXW400X package. Use of alternative power adapter may permanently damage the unit.

WARRANTY

Grandstream has a reseller agreement with our reseller customers. End users should contact the company from whom the product was purchased, for replacement, repair or refund.

If you purchased the product directly from Grandstream, contact your Grandstream Sales and Service Representative for an RMA (Return Materials Authorization) number. Grandstream reserves the right to change the warranty policy without prior notification.

Caution: *Changes or modifications to this product not expressly approved by Grandstream, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty.*

- This document is contains links to Grandstream GUI Interfaces. Please download these examples http://www.grandstream.com/products/gxw_series/gxw400x/documents/gxw400x_gui.zip for your reference.

- This document is subject to change without notice. The latest electronic version of this user manual is located at http://www.grandstream.com/products/gxw_series/gxw400x/documents/gxw400x_usermanual_english.pdf

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CONNECT YOUR GXW400X GATEWAY

Connecting the GXW400X gateway is easy. Before you begin, please verify the contents of the GXW400X package.

EQUIPMENT PACKAGING

Unpack and check all accessories. Equipment includes:

- one device unit
- one RJ45 Ethernet cable
- one 12V universal power adapter

CONNECT THE GXW400X

Follow these four (4) steps to connect your GXW400X gateway to the Internet and access the unit's configuration pages.

1. Connect standard touch-tone analog phones to the FXS ports.
2. Insert an RJ45 Ethernet cable into the WAN port of GXW400X and connect the other end to an uplink port (a router or a modem, etc.) Note: If the gateway is used in router mode, connect a PC to the LAN port of GXW400X for initial configuration.
3. Plug the power adapter into the GXW400X gateway into a power outlet.

Follow the instructions from the topic "Configuring GXW 400X with Web Browser" for initial configuration. The GUI pages will guide you through the remaining steps to set-up your gateway. Examples of the GUI Interfaces can be downloaded from: http://www.grandstream.com/products/gxw_series/gxw400x/documents/gxw400x_gui.zip.

GXW 400x

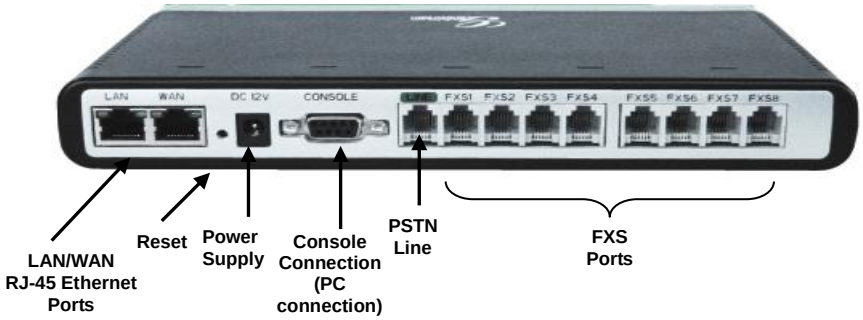

FIGURE 1: DIAGRAM OF GXW4004-8 BACK PANEL

TABLE 1: DEFINITIONS OF THE GXW CONNECTORS

LAN	Connect the LAN port with an Ethernet cable to your PC.
WAN	Connect to the internal LAN network or router.
RESET	Factory Reset button. Press for 7 seconds to reset factory default settings.
DC 12V	Power adapter connection
CONSOLE	Console connection for a PC.
LINE	When the unit loses power or unit became unregistered, FXS port 1 will be able to make/receive calls from the PSTN line connected to this port.
FXS1 – FXS8	FXS port to be connected to analog phones / fax machines.

Note: Once the GXW400X is turned on and configured, the front display panel indicates the status of the unit.

GXW4008

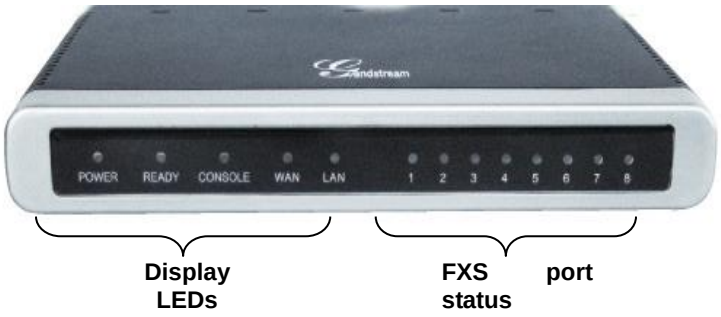

FIGURE 2: DIAGRAM OF GXW4004-8 DISPLAY PANEL

TABLE 2: DEFINITIONS OF THE GXW DISPLAY PANEL

Power LED	Indicates Power. Remains ON when Power is connected and turned ON.
Ready LED	Remains ON after boot-up.
Console	Turns ON when Console port is being used.
LAN LED	Indicates LAN (or WAN) port activity
WAN LED	Indicates PC (or LAN) port activity
LED 1–8	Indicate status of the respective FXS Ports on the back panel **Busy** - ON (Solid Green) **Available** - OFF Slow blinking FXS LEDs indicates Voice Mail for that port.

NOTE:

- All LED lights display green when ON.
- Slow blinking of READY, WAN and LAN LED together indicates a firmware upgrade or provisioning state.
- LED POWER, READY and WAN lights are ON when device is up and running and successfully registered to the SIP Server.
- Slow blinking of READY LED indicates that device has not registered with any SIP Service provider.

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GXW400X FEATURES

The GXW4000 series is a next generation IP voice gateway that is interoperable and compatible with leading IP-PBXs, Softswitches and SIP platforms. The GXW4000 series FXS gateway is auto-configurable, remotely manageable and scalable. The GXW4000 series gateways come in two models - the GXW-4004 and GXW-4008, each offering superb voice quality, traditional telephony functionality, easy deployment, and 4 and 8 FXS ports respectively. Each model features flexible dialing plans, integrated call routing to support a pure IP network call and an external power supply.

SOFTWARE FEATURES OVERVIEW

- 4 or 8 FXS ports
- Two RJ-45 ports (switched or routed)
- Multiple SIP accounts & profiles (4,8 / configurable through 2 profiles)
- Supports Voice Codecs:

  G711(a/μ, Annex I & II), G723.1A, G726 (ADPCM with 16/24/32/40 bit rates), G729 A/B/E, iLBC
- T.38 Fax
- Comprehensive Dial Plan support for Outgoing calls.
- G.168 Echo Cancellation
- Voice Activation Detection (VAD), Comfort Noise Generation (CNG), and Packet Loss Concealment (PLC)
- Supports PSTN/PBX analog telephone sets or analog trunks

TABLE 3: GXW400X SOFTWARE FEATURES

	GXW4004	GXW4008
Telephone Interfaces	4 FXS ports	8 FXS ports
SIP Provisioning	4 SIP accounts, 2 profiles	8 SIP accounts, 2 profiles
Network Interface	Two (2) 10M/100 Mbps, RJ-45	Two (2) 10M/100 Mbps, RJ-45
Number of Concurrent Calls (except when using SRTP)	4 Concurrent Calls	8 Concurrent Calls
Voice over Packet Capabilities	Voice Activity Detection (VAD) with CNG (comfort noise generation) and	Voice Activity Detection (VAD) with CNG (comfort noise generation) and PLC

	PLC (packet loss concealment), AEC with NLP Packetized Voice Protocol Unit (supports RTP and AAL2 protocol), G.168 compliant Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711	(packet loss concealment), AEC with NLP Packetized Voice Protocol Unit (supports RTP and AAL2 protocol), G.168 compliant Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711
PSTN Fail-over	PSTN failover port on power failure	PSTN failover port on power failure
Voice Compression	G.711 + Annex I (PLC), Annex II (VAD/CNG format) encoder and decoder, G.723.1A, G.726(ADPCM with 16/24/32/40 bit rates), G.729A/B/E, iLBC G.726 provides proprietary VAD, CNG, and signal power estimation, Voice Play Out unit (reordering, fixed and adaptive jitter buffer, clock synchronization), AGC (automatic gain control), Status output, Decoder controlling via voice packet header	G.711 + Annex I (PLC), Annex II (VAD/CNG format) encoder and decoder, G.723.1A, G.726(ADPCM with 16/24/32/40 bit rates), G.729A/B/E, iLBC G.726 provides proprietary VAD, CNG, and signal power estimation, Voice Play Out unit (reordering, fixed and adaptive jitter buffer, clock synchronization), AGC (automatic gain control), Status output, Decoder controlling via voice packet header
DHCP Server/Client	Yes, NAT Router or Switched Mode	Yes, NAT Router or Switched Mode
Fax over IP	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through, Fax Datapump V.17, V.19, V.27ter, V.29 for T.38 fax relay	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through, Fax Datapump V.17, V.19, V.27ter, V.29 for T.38 fax relay
QoS	Diffserv, TOS, 802.1 P/Q VLAN tagging	Diffserv, TOS, 802.1 P/Q VLAN tagging
Transport Protocol	RTP	RTP
DTMF Method	In-audio, RFC2833, and/or SIP Info	In-audio, RFC2833, and/or SIP Info
IP Signaling	SIP (RFC 3261)	SIP (RFC 3261)
Provisioning	TFTP, HTTP, HTTPS	TFTP, HTTP, HTTPS
Control	TLS/SIPS	TLS/SIPS
Management	Syslog support, HTTP and Telnet access	Syslog support, HTTP and Telnet access
Dial Plan	Yes	Yes

UPnP Support	Yes	Yes
Caller ID	Bellcore Type 1 & 2, ETSI, BT, NTT, and DTMF-based CID	Bellcore Type 1 & 2, ETSI, BT, NTT, and DTMF-based CID
Polarity Reversal / Wink	Yes	Yes
EMC	EN55022/EN55024 and FCC part15 Class B	EN55022/EN55024 and FCC part15 Class B

HARDWARE SPECIFICATION

The hardware specifications of the GXW FXS series are detailed in Table 4.

TABLE 4: HARDWARE SPECIFICATION OF GXW400X SERIES GATEWAYS

	GXW4004	GXW4008
Ports	4 FXS Ports	8 FXS Ports
Network interface	1 RJ45 for LAN, 10/100 Base-TX, Full Duplex 1 RJ45 for WAN,10/100Base-TX, Full Duplex	1 RJ45 for LAN, 10/100 Base-TX, Full Duplex 1 RJ45 for WAN,10/100Base-TX, Full Duplex
PSTN Port	PSTN fail-over port	PSTN fail-over port
Console	DB9 Serial port	DB9 Serial port
Universal Switching Power Adaptor	**Input**: 100-240V AC, 50/60Hz, 0.3A Max **Output**: 12V DC, 1.25A **UL** certified	**Input**: 100-240V AC, 50/60Hz, 0.3A Max **Output**: 12V DC, 1.25A **UL** certified
Short Haul Loop	2REN: Up to 1km on 24 AWG wire	2REN: Up to 1km on 24 AWG wire
Mounting	Wall mount, Desktop	Wall mount, Desktop
Dimension	225mm (L) x 135mm (W) x 35mm (H)	225mm (L) x 135mm (W) x 35mm (H)
Weight	0.29 lbs (3.5 oz)	0.29 lbs (3.5 oz)
Temperature	32~104°F / 0~40°C	32~104°F / 0~40°C
Humidity	10% - 90% (non-condensing)	10% - 90% (non-condensing)
Safety	UL	UL
Compliance	FCC, CE	FCC, CE

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BASIC OPERATIONS

UNDERSTANDING GXW VOICE PROMPTS

GXW400X has a built-in voice prompt menu for simple device configuration. To enter the voice prompt menu, **press ***** on the standard analog phone connected to any FXS port.

TABLE 5: DEFINITIONS OF THE GXW VOICE PROMPTS

Menu	Voice Prompt	User's Options
Main Menu	"Enter a Menu Option"	Enter "*" for the next menu option Enter "#" to return to the main menu Enter 01 – 05, 07,10 - 17, 47, 86 or 99 Menu option
01	"DHCP Mode", "PPPoE Mode "or "Static IP Mode"	Enter '9' to toggle the selection *If user selects "Static IP Mode", user need configure all the IP address information through menu 02 to 05. If user selects "Dynamic IP Mode", the device will retrieve all IP address information from DHCP server automatically when user reboots the device.*
02	"IP Address " + IP address	The current WAN IP address is announced Enter 12-digit new IP address if in Static IP Mode.
03	"Subnet " + IP address	Same as Menu option 02
04	"Gateway " + IP address	Same as Menu option 02
05	"DNS Server " + IP address	Same as Menu option 02
07	Preferred Vocoder	Enter "9" to go to the next selection in the list: • PCM U • PCM A • iLBC • G-726 • G-723 • G-729
10	"MAC Address"	Announces the Mac address of the unit.
12	WAN Port Web Access	Enter "9" to toggle between **enabled** and **disabled**
13	Firmware Server IP Address	Announces current Firmware Server IP address. Enter 12 digit new IP address.
14	Configuration Server IP Address	Announces current Config Server Path IP address. Enter 12 digit new IP address.

15	Upgrade Protocol	Upgrade protocol for firmware and configuration update. Enter "9" to toggle between **TFTP** and **HTTP**
16	Firmware Version	Firmware version information.
17	Firmware Upgrade	Firmware upgrade mode. Enter "9" to rotate among the following three options: 1. always check 2. check when pre/suffix changes 3. never upgrade
47	"Direct IP Calling"	Enter the target IP address to make a direct IP call, after dial tone. (See "*Make a Direct IP Call*".)
71-78	Phone calls between different ports of the same GW 400x	Applies to GXW4004/GXW4008. User can make internal calls between different ports inside the same GW even without being registered to SIP server. By dialing *** and 7x user can reach colleague connected to port "x".
86	Voice Mail	Number of voice mails
99	"RESET"	Enter "9" to reboot the device; or Enter MAC address to restore factory default setting (See **Restore Factory Default Setting** section)
	"Invalid Entry"	Automatically returns to Main Menu

Five Success Tips when using the Voice Prompt

1. "*" shifts down to the next menu option
2. "#" returns to the main menu
3. "9" functions as the ENTER key in many cases to confirm an option
4. All entered digit sequences have known lengths - 2 digits for menu option and 12 digits for IP address. For IP address, add **0** before the digits if the digits are less than 3 (i.e. - 192.168.0.26 should be key in like 192168**00**0**0**26. No decimal is needed).
5. Key entry cannot be deleted but the phone may prompt error once it is detected

PLACING A PHONE CALL

PHONE OR EXTENSION NUMBERS

1. Dial the number directly and wait for 4 seconds (To change the default value, modify the following setting - "No Key Entry Timeout")

or

2. Dial the number directly and press # (Use # as dial key" must be configured in web configuration).

Examples:

1. Dial a number (e.g. (626) 666-7890), first enter the prefix number (usually 1+ or international code) followed by the phone number. Press # or wait for 4 seconds. Check with your VoIP service provider for further details on prefix numbers.

DIRECT IP CALLS

Direct IP calling allows two parties, that is, a FXS Port with an analog phone and another VoIP Device, to talk to each other in an ad hoc fashion without a SIP proxy.

Elements necessary to completing a Direct IP Call:

1. Both GXW400X and other VoIP Device, have public IP addresses, or
2. Both GXW400X and other VoIP Device are on the same LAN using private IP addresses, or
3. Both GXW400X and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

GXW400X supports two ways to make Direct IP Calling:

Using IVR

1. Pick up the analog phone then access the voice menu prompt by dial "*******"
2. Dial "**47**" to access the direct IP call menu
3. Enter the IP address using format ex. **192*168*0*160** after the dial tone.

Using Star Code

1. Pick up the analog phone then dial "***47**"
2. Enter the target IP address using same format as above.

Note: **NO** dial tone will be played between step 1 and 2.

Destination ports can be specified by using "*****" (encoding for ":") followed by the port number.

Examples:

a) If the target IP address is 192.168.0.160, the dialing convention is
***47 or Voice Prompt with option 47, then 192*168*0*160.**
Followed by pressing the "**#**" key if it is configured as a send key or wait 4 seconds. In this case, the default destination port 5060 is used if no port is specified.

b) If the target IP address/port is 192.168.1.20:5062, then the dialing convention would be: ***47 or Voice Prompt with option 47, then 192*168*0*160*5062** followed by pressing the "#" key if it is configured as a send key or wait for 4 seconds.

NOTE: When completing direct IP call, the "**Use Random Port**" should set to "NO". You can not make direct IP calls between FXS ports since they are using same IP.

CALL HOLD

Place a call on hold by pressing the "flash" button on the analog phone (if the phone has that button). Press the "flash" button again to release the previously held Caller and resume conversation. If no "flash" button is available, use "hook flash" (toggle on-off hook quickly). You may drop a call using hook flash.

CALL WAITING

Call waiting tone (2 short beeps) indicates an incoming call, if the call waiting feature is enabled. Toggle between incoming call and current call by pressing the "flash" button. First call is placed on hold. Press the "flash" button to toggle between two active calls.

CALL TRANSFER

BLIND TRANSFER

Assume that call Caller A and B are in conversation. A wants to *Blind Transfer* B to C:

3. Caller A presses **FLASH** on the analog phone to hear the dial tone.
4. Caller A dials ***87** then dials caller C's number, and then **#** (or wait for 4 seconds).
5. Caller A will hear the dial tone. Then, A can hang up.

NOTE: "*Enable Call Feature*" must be set to "Yes" in web configuration page.

ATTENDED TRANSFER

Assume that Caller A and B are in conversation. Caller A wants to *Attend Transfer* B to C:

1. Caller A presses **FLASH** on the analog phone for dial tone.
2. Caller A then dials Caller C's number followed by # (or wait for 4 seconds).

3. If Caller C answers the call, Caller A and Caller C are in conversation. Then A can hang up to complete transfer.
4. If Caller C does not answer the call, Caller A can press “flash” to resume call with Caller B.

NOTE: When Attended Transfer fails and A hangs up, the GXW400X will ring back user A to remind A that B is still on the call. A can pick up the phone to resume conversation with B.

3-WAY CONFERENCING

The GXW400X supports Bellcore style 3-way Conference.

Instructions for 3-way conference:

Assuming that call party A and B are in conversation. A (GXW400X) wants to bring C in a conference:

1. A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone.
2. A dials C’s number then # (or wait for 4 seconds).
3. If C answers the call, then A presses FLASH to bring B, C in the conference.
4. If C does not answer the call, A can press FLASH back to talk to B.
5. If A presses FLASH during conference, C will be dropped out.
6. If A hangs up, the conference will be terminated for all three parties when configuration “Transfer on Conference Hangup” is set to “No”. If the configuration is set to “Yes”, A will transfer B to C so that B and C can continue the conversation.

HUNTING GROUP

This feature allows the user to setup a single SIP account on the gateway and have the ability to use all FXS ports to make/receive calls. Using this feature, all ports active in same Hunting Group will have the same phone number and incoming calls will be distributed in a Linear or Circular manner among the ports active in that Hunting Group. The number of hunting groups is limited by the number of ports each GXW model has - i.e. each port can be its own Hunting Group. The most practical and efficient way to use Hunting Groups is to assign 2 or 3 ports to separate Hunting Groups.

One additional and popular way to use the Hunting Group feature is called “*multiplexed analog lines*”. In this configuration, a legacy PBX system with 8 FXO trunks can be connected to 8 GXW 4008 ports

configured as Hunting Group. The GXW can be registered to a SIP server provider using only one phone number. If the SIP service provider allows multiple calls to the same number, the GXW will allow 8 concurrent calls to the same SIP number. All office members can be reached remotely using the same phone number in a round-robin fashion.

Example Configuration of a typical Hunting Group:

1. Configure the SIP account from your VoIP Service Provider on **FXS port 1** under **FXS Ports** webpage.
2. Select **Active** under the **Hunting Group** drop box for FXS port 1.
3. For the remaining ports (say 2, 3 and 4) select **1** for **Hunting Group**. Ports 2, 3 and 4 are now active members of the hunting group associated with port 1.

This configuration will route all calls directed to FXS port 1 to ports 2, 3 and/or 4 in round robin fashion respectively *if* port 1 is busy or times out. You can configure the ring timeout on the **Profile** page.

Example configuration of a multiple Hunting Group:

FXS Port #1: SIP UserID and Authenticate ID entered, Hunting group set to "**Active**"

FXS Port #2: SIP UserID and Authenticate ID left blank, Hunting Group set to "**1**"

FXS Port #3: SIP UserID and Authenticate ID left blank, Hunting Group set to "**1**"

FXS Port #4: SIP UserID and Authenticate ID entered, Hunting group set to "**Active**"

FXS Port #5: SIP UserID and Authenticate ID left blank, Hunting Group set to "**4**"

FXS Port #6: SIP UserID and Authenticate ID left blank, Hunting Group set to "**4**"

FXS Port #7: SIP UserID and Authenticate ID entered, Hunting group set to "**Active**"

FXS Port #8: SIP UserID and Authenticate ID left blank, Hunting Group set to "**7**"

Hunting Group 1 contains ports 1, 2, 3. Hunting Group 4 contains ports 4, 5, 6.
Hunting Group 7 contains ports 7, 8.

Please be aware, the choice of 1 for ports 2 and 3, the choice of 4 for ports 5 and 6, the choice 7 for port 8 is required to indicate that the SIP account tied to port marked as **"Active"** will be used for all members of the same Hunting group. Needless to say, those members of the same Hunting group may not be sequential ports. In following example ports 3, 5 and 7 tied to SIP Account configured in Port #1 marked as **"Active"**, and ports 4,6,8 tied to SIP Account configured in Port #2 marked as **"Active"** as well.

Example of not sequential configuration of a multiple Hunting Group:

FXS Port #1: SIP UserID and Authenticate ID entered, Hunting group set to "**Active**"

FXS Port #2: SIP UserID and Authenticate ID entered, Hunting Group set to "**Active**"

FXS Port #3: SIP UserID and Authenticate ID left blank, Hunting Group set to "**1**"

FXS Port #4: SIP UserID and Authenticate ID left blank, Hunting group set to "**2**"

FXS Port #5: SIP UserID and Authenticate ID left blank, Hunting Group set to "**1**"

FXS Port #6: SIP UserID and Authenticate ID left blank, Hunting Group set to "**2**"

FXS Port #7: SIP UserID and Authenticate ID left blank, Hunting group set to "**1**"

FXS Port #8: SIP UserID and Authenticate ID left blank, Hunting Group set to "**2**"

Note: A single call directed to the SIP account will NOT result in all ports ringing at the same time. They will ring in the hunting group only. This feature is applicable to incoming calls only.

There are two types of hunting groups, Linear and Circular. Linear style will sort the call to the lowest-numbered available line, this is also called “serial hunting”. Circular style will distribute the calls "round-robin". If a call is assigned to line 1, the next call goes to 2 and the next to 3. The succession throughout each of the lines continues even if one of the previous lines becomes available. When the end of the hunt group is reached, the hunting starts over at the first line. Lines are skipped if they are still busy on a previous call. These two hunting styles can be configured from the Profile_x page.

INTER-PORT CALLING

In some cases a user may want to make phone calls between the phones connected to multiple ports of the same gateway when it is used as a stand alone unit, without the use of a SIP server. This feature will also be applicable when the gateway is used with Hunting Groups and is registered to SIP server only with one master number. In such cases users still will be able to make inter-port calls by using the IVR feature.

For example on the GX4004 and GXW4008 the user connected to port 1 can reach the user connected to port 3 by dialing *** and 73. Digit 7 indicated using inter-port calling feature, digit 3 indicates the port number which should be reached. At the same manner the user connected to port 4 can reach the user connected to port 8 by dialing *** and 78.

PSTN PASS THROUGH/LIFE LINE

The RJ-11 line jack on the GXW400X side connected to a legacy PSTN line, functions as a pass through jack when the GXW400X loses power or becomes unregistered. In this case, analog phone connected to FXS port 1 will be directly connected to RJ-11 jack marked as LINE. The pass through/life line mode enables the user to use the analog phone for PSTN calls directly without using an access code.

SENDING AND RECEIVING FAX

GXW400X supports fax in two modes: 1) T.38 (Fax over IP) and 2) Fax Pass through. T.38 is the preferred method because it is more reliable and works well in most network conditions. If the service provider supports T.38, please use this method by selecting T.38 as fax mode (default). If the service provider does not support T.38, pass-through mode may be used. If you have problems with sending or receiving Fax, toggle the Fax Tone Detection Mode setting.

SUPPORT RADIUS PROTOCOL

GXW400X supports RADIUS for authentication, authorization and billing purposes. Primary and secondary RADIUS server configurations are available to provide redundancy to this feature. In case Primary Radius server becomes unusable, RADIUS requests will be automatically sent to the secondary server. When at least one RADUIS server was configured, the device will allow users to make phone calls only after authorization from RADIUS server has been received. CDR (Call Detail Record) is also sent to the RADIUS server for billing purposes. RAIDUS server can send requests to terminate calls when run out of pre-paid credit.

The GXW400X will be able to work in VoIP billing environment using redundant double server configuration. User will be able to configure primary and secondary RADUIS server IP Addresses or FQDNs. Once at least one RADUIS server was configured, the device will allow users to make phone calls only after permission from RADIUS server has been received. In case Primary Radius server becomes unusable, secondary will take role of primary and will manage credit recourses in the network. Imbedded RADIUS client also supports request generated by Radius server to terminate calls when run out of pre-paid credit.

CALL FEATURES

GXW400X supports the traditional telephony features available in a PBX as well as additional advanced telephony features.

TABLE 6: CALL FEATURES TABLE (STAR CODE)

Key	Call Features
***02**	**Forcing a Codec** (per call) *027110 (PCMU), *027111 (PCMA), *02723 (G723), *02729 (G729), *0272616 (G726-r16), *0272624 (G724-r24), *0272632 (G726-r32), *0272640 (G726-r40), *027201 (iLBC)
***03**	**Disable LEC** (pe call) Dial "*03" + " number ". **No** dial tone is played in the middle.
***16**	**Enable SRTP**
***17**	**Disable SRTP**
***30**	**Block CallerID** (for all-config change)
***31**	**Send CallerID** (for all-config change)
***67**	**Block CallerID** (per call)
***82**	**Send CallerID** (per call)
***47**	**Direct IP Calling.** Dial "*47" + "IP address". **No** dial tone will be played in the middle. Detail see Direct IP Calling section on page 12.
***50**	**Disable Call Waiting** (for all-config change)
***51**	**Enable Call Waiting** (for all-config change)
***69**	**Call Return Service:** Dial *69 and the phone will dial the last incoming phone number received.
***70**	**Disable Call Waiting** (Per Call)
***71**	**Enable Call Waiting** (Per Call)
***72**	**Unconditional Call Forward:** Dial "*72" and then the forwarding number followed by "#". Wait for dial tone and hang up. (dial tone indicates successful forward)
***73**	**Cancel Unconditional Call Forward:** Dial "*73" and wait for dial tone, then hang up.
***74**	**Enable Paging Call:** Dial "*74" and then the destination phone number you want to activate in Paging mode.
***78**	**Enable Do Not Disturb (DND):** When enabled all incoming calls will be rejected.
***79**	**Disable Do Not Disturb (DND):** When disabled, incoming calls will be accepted.

***87**	**Blind Transfer**
***90**	**Busy Call Forward:** Dial "*90" and then the forwarding number followed by "#". Wait for dial tone then hang up.
***91**	**Cancel Busy Call Forward**: dial "*91". Wait for dial tone. Hang up.
***92**	**Delayed Call Forward:** Dial "*92" and then the forwarding number followed by "#". Wait for dial tone then hang up.
***93**	**Cancel Delayed Call Forward:** Dial "*93" for a dial tone, then hang up.
Flash/Hook	If user hears call waiting beep, flash/hook will switch to the new incoming call. Also used to switch to a new channel for a new call.
#	Pressing pound sign will serve as **Re-Dial** key.

CONFIGURATION GUIDE

CONFIGURING GXW400X VIA VOICE PROMPT

DHCP MODE

Select voice menu option 01 to enable GXW400X to use DHCP.

STATIC IP MODE

Select voice menu option 01 to enable GXW400X to use STATIC IP mode, then use option 02, 03, 04, 05 to set up IP address, Subnet Mask, Gateway and DNS server respectively.

FIRMWARE SERVER IP ADDRESS

Select voice menu option 13 to configure the IP address of the firmware server.

CONFIGURATION SERVER IP ADDRESS

Select voice menu option 14 to configure the IP address of the configuration server.

UPGRADE PROTOCOL

Select voice menu option 15 to choose firmware and configuration upgrade protocol. User can choose between TFTP and HTTP.

FIRMWARE UPGRADE MODE

Select voice menu option 17 to choose firmware upgrade mode among the following three options:

1) always check, 2) check when pre/suffix changes, and 3) never upgrade

WAN PORT WEB ACCESS

Select voice menu option 12 to enable WAN Port Wed Access of the device configuration pages.

CONFIGURING GXW400X WITH WEB BROWSER

The GXW400X series gateway has an embedded Web server that allows users to configure the GXW400X through a web browser.

ACCESS THE WEB CONFIGURATION MENU

The GXW400X HTML configuration menu can be accessed via LAN or WAN port:

From the LAN port (For GXW4004/8 only)::

1. Directly connect a computer to the LAN port.
2. In Windows, open a 'Command Prompt' window
3. Type in "ipconfig /release", the IP address etc. becomes 0.
4. Type in "ipconfig /renew", the computer gets an IP address in 192.168.2.x segment by default
5. Open a web browser, type in the default gateway IP address. http://192.168.2.1. You will see the login page of the device.

From the WAN port (For GXW4004/8 only):

The WAN port HTML configuration option is disabled by default from factory. To access the HTML configuration menu from the WAN port:

1. Follow table 4 to find the WAN side IP address.
2. Open a web browser, type in the WAN side IP address – for example: http://GXW400X -IP-Address (the GXW400X IP-Address is the WAN IP address for the GXW400X).

NOTE:

- WAN side HTTP access is disabled by default for security reasons. You can enable HTTP access on the configuration page by setting "*WAN side HTTP access*" to be **YES**.
- Initial access to the configuration pages is always from the LAN port. The instructions are listed above.
- The IVR announces 12 digits IP address, you need to strip out the leading "0" in the IP address. For ex. IP address: 192.168.001.014, you need to type in http://192.168.1.14 in the web browser.

Once the HTTP request is entered and sent from a web browser, the user will see a log-in screen. There are two default passwords for the login page:

User Level:	Password:	Web pages allowed:
End User Level	123	Only Status and Basic Settings
Administrator Level	Admin	Browse all pages

The password is case sensitive with maximum length of 25 characters. The factory default password for End User and administrator is "123" and "admin" respectively. Only an administrator can access the "ADVANCED SETTING", "Profile 1", "Profile 2" and "FXS PORTs" configuration pages. Please reference the GUI pages using the following link: http://www.grandstream.com/products/gxw_series/gxw400x/documents/gxw400x_gui.zip .

IMPORTANT SETTINGS

The end-user must configure the following settings according to the local environment.

NOTE: Most settings on the web configuration pages are set to the default values.

NAT SETTINGS

If you plan to keep the gateway within a *private network* behind a firewall, we recommend using STUN Server. The following three (3) settings are useful in the STUN Server scenario:

1. **STUN Server** (under Advanced Settings webpage)
   Enter a STUN Server IP (or FQDN) that you may have, or look up a free public STUN Server on the internet and enter it on this field. If using Public IP, keep this field blank.
2. **Use Random Ports** (under Advanced Settings webpage)
   It really depends on your network settings, so set this parameter to Yes or No, whichever works. Generally if you have multiple IP devices under the same network, it should be set to Yes. If using a Public IP address, set this parameter to No.
3. **NAT Traversal** (under the Profile web pages)
   Set this to Yes when gateway is behind firewall on a private network.

DTMF METHODS

DTMF Settings are in Profile pages.

- DTMF in-audio
- DTMF via RTP (RFC2833)
- DTMF via SIP INFO

You can enable set priority of DTMF methods according to your preference, from Priority 1 to 3. This setting should be based on your server DTMF setting.

PREFERRED VOCODER (CODEC)

The GXW400X supports a broad range of voice codecs. Under Profile web pages, choose your preferred order of different codecs:

- PCMU/A (or G711μ/a)
- G729 A/B/E
- G723

- G726 (16/24/32/40)
- iLBC
- AAL2 (all G726)

DEFINITIONS

This section will describe the options in the Web configuration user interface. As mentioned, a user can log in as an administrator or end-user.

Functions available for the end-user are:

- **STATUS:** Displays the network status, account status, software version and MAC-address of the phone
- **BASIC SETTINGS:** Basic preferences such as date and time settings, multi-purpose keys and LCD settings can be set here.

Additional functions available to administrators are:

- **ADVANCED SETTINGS:** To set advanced network settings, codec settings and XML configuration settings.
- **PROFILE X:** To configure each of the SIP accounts.
- **FXS PORTS:** To configure each of the FXS ports and Hunting Groups etc.

TABLE 7: BASIC SETTINGS

End User Password	Password to access the Web Configuration Menu. This field is case sensitive with a maximum length of 25 characters.
Web Port	By default, HTTP uses port 80. This field is for customizable web port.
Telnet Server	Default is Yes.
IP Address	There are two modes to operate the GXW400X : **DHCP mode**: The GXW400X acquires its IP address from the first DHCP server it discovers from the LAN It is on. **PPPoE:** Set the PPPoE account settings. The GXW400X will establish a PPPoE session if any of the PPPoE fields is set. **Static IP mode:** configure the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields. These fields are set to zero by default.

DHCP hostname	This option specifies the name of the client. This field is optional but may be required by some Internet Service Providers. Default is blank.
DHCP vendor class ID	Used by clients and servers to exchange vendor-specific information. Default is HT500.
PPPoE account ID	PPPoE username. Necessary if ISP requires you to use a PPPoE (Point to Point Protocol over Ethernet) connection.
PPPoE password	PPPoE account password.
PPPoE Service Name	This field is optional. If your ISP uses a service name for the PPPoE connection, enter the service name here. Default is blank.
Time Zone	Controls how the date/time is displayed according to the specified time zone.
Self Defined Time Zone	The syntax is **std offset dst [offset],start[/time],end[/time]** Default is set to : **MTZ+6MDT+5,M3.2.0,M11.1.0** **MTZ+6MDT+5,** It indicates a time zone with 6 hours offset with 1 hour ahead which is the US central time. It is positive (+) if the local time zone is west of the Prime Meridian and negative (-) if it is east. **Prime Meridian** (*International or Greenwich Meridian*) **M3.2.0,M11.1.0** The 1st number indicates Month: 1,2,3,..,12 (for Jan, Feb, .., Dec) The 2nd number indicates the nth iteration of the weekday: (1st Sunday, 3rd Tuesday etc) The 3rd number indicates Weekday: 0,1, 2, ..,6(for Sun, Mon, Tue, .., Sat) Therefore, this example is the DST which starts from the second Sunday of March to the 1st Sunday of November.
Language	Default is English. Languages supported with voice prompt.
Device Mode	This parameter controls whether the device is working in NAT router mode or Bridge mode. Save the setting and reboot prior to configuring the GXW.
NAT maximum ports	Maximum NAT ports, typically one connection per port.
NAT TCP timeout	NAT TCP timeout in seconds
NAT UDP timeout	NAT UDP timeout in seconds.
Uplink Bandwidth	ISP specified WAN uplink bandwidth

Downlink Bandwidth	ISP specified WAN downlink bandwidth
Enable UPnP support	If set to “Yes”, the GXW400X would act as an UPnP gateway for your UPnP enable applications. **UPnP** - Universal Plug n Play
Reply to ICMP on WAN port	If set to “Yes”, the GXW400X will respond to the PING. Default is **No**. “Yes” setting may make the gateway vulnerable to a Denial of Service attack.
WAN side HTTP/Telnet Access	If set to “Yes”, user can access the configuration page through the WAN port, instead of through the “PC” port. WARNING: this configuration is less secure than default option. Default is **No**.
White list for WAN side	List the IP address or IP range in the White list Note for list Rules: HTTP/telnet: Case 1: WAN side telnet/HTTP access enabled - If white list exists, then ONLY these IP addresses are ALLOWED to web and telnet access. - If black list exists and white list is empty, then ONLY these IP addresses are NOT ALLOWED to web and telnet access. Case 2: WAN side telnet/HTTP access is not allowed: -All addresses are NOT ALLOWED http and telnet access. SIP/RTP: - If both white list and black list are empty, the behaviour is same as before. - If white list exists, then ONLY these IP addresses are ALLOWED SIP/RTP access. - If black list exists and white list is empty, then ONLY these IP addresses are NOT ALLOWED SIP/RTP access.
Black list for WAN side	List the IP address or IP range in the White list. The same rules as white list.
Cloned WAN MAC Address	This allows you to change/set the MAC address on the WAN interface.
Enable LAN DHCP	Default is Yes.
LAN Subnet Mask	Sets the LAN subnet mask. Default value is 255.255.255.0
LAN DHCP Base IP	Base IP for the LAN port which functions as a Gateway for the subnet. Default value is **192.168.2.1**. When the device detects WAN IP is conflicting with LAN IP, the LAN base IP address will be changed based on the network mask -- the effective subnet will be increased by 1. For example; 192.168.2.1 will be changed to 192.168.3.1 if net mask is 255.255.255.0. Then the device will reboot

LAN DHCP Start IP	Default is 100
LAN DHCP End IP	Default is 199
DHCP IP Lease Time	Value is set in units of hours. Default value is **120 hrs** (5 Days.) The time IP address is assigned to the LAN clients.
DMZ IP	Forward all WAN IP traffic to a specific IP address if no matching port is used by GXW400X or defined in port forwarding.
Port Forwarding	Forwards a matching (TCP/UDP) port to a specific LAN IP address with a specific (TCP/UDP) port.

In addition to the Basic Settings configuration page, end users also have access to the Device Status page.

TABLE 8: STATUS PAGE

MAC Address	The device ID in hexadecimal format. This is needed for Internet Service Provider troubleshooting. **Note: there are separate MAC addresses for the WAN side and the LAN side.** The LAN MAC address will be used for provisioning and can be found on the label on original box and on the label located on the bottom panel of the device.
WAN IP Address	Shows WAN IP address of GXW400X
Product Model	Contains the product model info.
Software Version	Program: This is the main software release. Boot and Loader are not changed often.
System Up Time	Shows system uptime since the last reboot.
PPPoE Link Up	Shows whether the PPPoE connection is running if connected to DSL modem.
NAT	Shows type of NAT the GXW400X is connected to via its WAN port. It is based on STUN protocol.

Port Status	Displays relevant information regarding the individual FXS ports. Example:

Port	Hook	Registration	DND	Forward	Busy Forward	Delayed Forward
FXS1	**On Hook**	**Registered**	No	613		
FXS2	**Off Hook**	**Registered**	No		614	
FXS3	**On Hook**	**Not Registered**	No			
FXS8	**On Hook**	**Registered**	Yes			615

** FXS port 8 user has set Do Not Disturb.
FXS port 1 user has set his calls to be forwarded unconditionally to ext 613
FXS port 2 user has set his calls to be forwarded to 614 when his phone is busy.
FXS port 3 user is not registered with his SIP Server.

Advanced User configuration includes not only the end user configuration, but also advanced configurations such as: SIP configuration, Codec selection, NAT Traversal Setting and other miscellaneous configuration.

TABLE 9: ADVANCED SETTINGS

Admin Password	Administrator password. Only the administrator can configure the "Advanced Settings" page. Password field is purposely left blank for security reasons after clicking update and saved. The maximum password length is 25 characters.
Layer 3 QoS	This field defines the layer 3 QoS parameter which can be the value used for IP Precedence or Diff-Serv or MPLS. Default value is **48**.
Layer 2 QoS	Value used for layer 2 VLAN tag. Default setting is **blank**.
STUN Server	IP address or Domain name of the STUN server. STUN server assists devices behind a NAT firewall or a router to set up calls via VoIP. STUN server is often provided by the VoIP service provider or a public STUN server can be used. For a sample list of public free STUN servers please refer to: http://www.voip-info.org/wiki-STUN
Keep-alive interval	This parameter specifies the number of seconds after which a new blank UDP packet will be sent out to the proxy/server port in order to have the port stay open and the device remain reachable. Default is **20 seconds**.
Use STUN to detect network connectivity	Use STUN keep-alive to detect WAN side network problems. If keep-alive request does not yield any response for configured number of times, the device will restart the TCP/IP stack. If the STUN server does not respond when the device boots up, the feature is disabled.
Firmware Upgrade & Provisioning	Enables GXW400X to download firmware or configuration file using either the TFTP or HTTP/S protocols.
Via TFTP Server	This is the IP address of the configured TFTP server. If selected and it is non-zero or not blank, the GXW400X retrieves the new configuration file or new code image from the specified TFTP server at boot time. After 5 attempts, the system will timeout and then it will start the boot process using the existing code image in the Flash memory. If a TFTP server is configured and a new code image is retrieved, the new downloaded image is saved into the Flash memory. **Note: Do NOT interrupt the firmware upgrade process (especially the power supply) as this will damage the device**. Depending on the network environment this process may take up to 15 or 20 minutes.

Via HTTP or HTTPS Server	The URL for the HTTP server used for firmware upgrade and configuration via HTTP. **Note:** If Auto Upgrade is set to No, GXW400X will only do HTTP download once at boot up.
Firmware Server Path	IP address or domain name of firmware server. That URL of the server that hosts the firmware release. The default server is: **fm.grandstream.com/gs**
Config Server Path	IP address or domain name of configuration server. The server hosts a copy of the configuration file to be installed on the gateway. The default server is: fm.grandstream.com/gs
XML Config File Password	The password used for encrypting the XML configuration file using OpenSSL. This is required for the phone to decrypt the encrypted XML configuration file.
HTTP/HTTPS User Name	The user name needed to authenticate withthe HTTP/HTTPS server.
HTTP/HTTPS Password	The password needed to authenticate with the HTTP/HTTPS server.
Always send HTTP Basic Authentication Information	Default is **Send HTTP Basic Authentication Information only when challenged**. If set to **Always**, device will send configured user name and password within HTTP request without server sending authentication challenge.
Firmware File Prefix	This field enables user to store different versions of firmware files in one single directory on the firmware server. If configured, only the firmware file with the matching prefix will be downloaded.
Firmware File Postfix	This field enables user to store different versions of firmware files in one single directory on the firmware server. If configured, only the firmware file with the matching postfix will be downloaded.
Config File Prefix	This field enables user to store different configuration files in one single directory on the configuration server. If configured, only the configuration file with the matching prefix will be downloaded.
Config File Postfix	This field enables user to store different configuration files in one single directory on the configuration server. If configured, only the configuration file with the matching postfix will be downloaded.
Allow DHCP Option 66 or 160 to override server	If set to “Yes”, configuration and upgrade server information can be obtained using DHCP option 66 or option 160 from DHCP server located in customer’s environment. **Note:** If DHCP Option 66 is enabled, the gateway will attempt downloading a configuration file from the server URL provided by DHCP, even though **Config Server Path** is left blank.

Automatic Upgrade	Choose "*Yes*" to enable automatic upgrade and provisioning. When set to No, GXW400X will only do upgrade once at boot up. When "Check every day" or "Check every week" is checked, user can specify "Hour of the day (0-23)" or "Day of the week (0-6)". Default time is Monday 1AM. There are three options to choose from: "Always check for New Firmware at Boot up", "Check New Firmware only when F/W pre/suffix changes", and "Always Skip the Firmware Check".
Disable SIP NOTIFY Authentication	Device will not challenge NOTIFY with 401 when set to Yes.
Authenticate Conf File	If set to Yes, configuration file is authenticated before being accepted. This protects the configuration from unauthorized modifications.
Firmware Key	For firmware encryption. It should be 32-digit in Hexadecimal Representation. End user should keep it blank.
SIP TLS Certificate	The GXW400X series supports SIP over TLS. It has built-in private key and SSL certificate. The user specified SSL certificate used for SIP over TLS is in X.509 format.
SIP TLS Private Key	You may also customize the SSL Private Key. The user specified SSL private key used for SIP over TLS is in X.509 format.
SIP TLS Private Key Password	Enter SSL Private Key password here.
ACS URL	User specify the Auto Configuration Server's URL (TR-069 protocol)
ACS Username	User specify the ACS Username
ACS Password	User specify the ACS password
Periodic Inform Enable	Default is No. If set to YES, device will send inform packets to the ACS
Periodic Inform Interval	Frequency that the inform packets will be sent out to the ACS
Connection Request Username	Set a user name for the ACS to connect to this device
Connection Request Password	Set a password for the ACS to connect to this device
Connection Request Port	Set a port number for the ACS to connect to this device, default is **7547**
CPE SSL Certificate	Configure the SSL authentication of Customer-premises equipment
CPE SSL Private Key	Configure the SSL Private Key of Customer-premises equipment

System Ring Cadence	Configuration option for all FXS ports ring cadence for all incoming calls. (Syntax: c=on1/off1-on2/off2-on3/off3; [...]) Default is set to c=2000/4000; (US standards)
Call Progress Tones	Using these settings, user can configure tone frequencies according to user preference. By default, the tones are set to North American frequencies. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds. ONis the period of ringing (ON time in ms) while OFF is the period of silence. In order to set a continuous ring, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern. • "Dial tone" • "Ringback tone" • "Busy/Re-order tone" • "Confirmation tone" Please refer the document below to determine your local call progress tones: http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf
Prompt Tone Access Code	Key pattern to get Prompt Access Tone. (Maximum 20 digits. No default)
Lock Keypad Update	If set to "Yes", the configuration update via keypad is disabled.
Disable Voice Prompt	Disables the voice prompt configuration. Default is "No". If set to "Yes" accessing integrated voice menu will be impossible.
Disable Direct IP Call	Disables the Direct IP Call function. Default is "No". If set to "Yes" direct IP-to-IP calling will not be supported.
Lifeline Mode	Life line feature ensures user can place/receive a PSTN call in an emergency situation. 1. If set to "**Auto**", in case of power loss or loss of SIP registration, the PSTN line will be seamlessly connected to analog phone connected to FXS port. 2. If set to "**Always Connected**" the PSTN line will be always connected to the phone connected to FXS port. VoIP calls will not be allowed in this configuration. 3. If set to "**Always Disconnected**", user can only place VoIP calls, regardless of any power loss and/or SIP registration problems. User will be unable to place/receive any PSTN calls.

Failover to FXO Gateway	This feature allows users to place an outbound PSTN call in case there is a loss of an active registration (SIP server unreachable) of all FXS profiles. If set to "YES", when GXW400X recognizes a loss of registration, all outgoing calls will be routed to an FXO gateway. The use of this option presumes a configured GXW410x or another FXO gateway with an active PSTN line connection.
FXO Gateway	IP Address or URI of the FXO gateway.
NTP server	URI or IP address of the NTP (Network Time Protocol) server. Used by the phone to synchronize the date and time. An extensive list of public NTP servers can be found at http://www.ntp.org
NTP Update Interval	Default is **1440**. Updates the Network Time Protocol (Values range from 5 – 1440 minutes)
Syslog Server	The IP address or URL of System log server. The server collects system log information from the device.
Syslog Level	Select the GXW400X to report the log level. Default is NONE. The level is one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events: 1. product model/version on boot up (INFO level) 2. NAT related info (INFO level) 3. sent or received SIP message (DEBUG level) 4. SIP message summary (INFO level) 5. inbound and outbound calls (INFO level) 6. registration status change (INFO level) 7. negotiated codec (INFO level) 8. Ethernet link up (INFO level) 9. SLIC chip exception (WARNING and ERROR levels) 10. memory exception (ERROR level) The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: GS_LOG: [device MAC address][error code] error message *Example*: May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up
Send SIP Log	If Syslog is enabled and Send SIP Log is set to YES, then SIP messages will also be delivered via Syslog. Default is set to NO.

Primary RADIUS Server	IP Address or FQDN of the primary RADIUS Server
Primary RADIUS Auth Port	Primary Radius server authentication port. Default value is 1812
Primary RADIUS Acct Port	Primary Radius server accounting port. Default value is 1813
Primary Radius Server Secret	Special secret string should be preconfigured according to RADIUS Server configuration
Secondary RADIUS Server	IP Address or FQDN of the secondary RADIUS Server
Secondary RADIUS Auth Port	Secondary Radius server authentication port. Default value is 1812
Secondary RADIUS Acct Port	Secondary Radius server accounting port. Default value is 1813
Secondary Radius Server Secret	Special secret string should be preconfigured according to RADIUS Server configuration
RADIUS Timeout	Default value is 2 seconds. The time between retries the GXW will send "Access-Request" message to RADIUS server in purpose to authenticate it.
RADIUS Retry	Default value is 3 times. Number of times the device will try to authenticate itself with preconfigured RADIUS server during initialization process.
Download Device Configuration	This setting allows user to download a text file containing all the P values of each setting as configured on the unit. (Note: For Security Reasons, any Password will not be Downloaded)
Download Device XML Configuration	Allows user to download and save a XML file containing all the P values of each setting as configured at that point on the unit. (Note: For Security Reasons, all Passwords won't be Downloaded)
Upload firmware	Allows user to upload firmware file from local directory.
Upload configuration	Allows user to upload configuration file from local directory.

TABLE 10: FXS PORTS SETTINGS

FXS Port	FXS Port Number
SIP User ID	User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.

Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.
Password	SIP service subscriber's account password for GXW400X to register to (SIP) servers of ITSP.
Name	Any name to identify this specific user.
Profile ID	Select the corresponding Profile ID (1/2)

Hunting Group	This feature enables the gateway to register all existing FXS ports with the same phone number. Each incoming call will be routed to first available port in Linear or Circular mode. User may configure all ports as members of the same Hunting Group or it may configure different port combinations for more than one Hunting Group. For example: Ports 1, 3 and 5 are members of the same Hunting Group, the rest of the ports may have separate numbers and may be reached independently. Any port, member of a Hunting Group that is not registered with a SIP account, will be able to place outbound calls using the SIP credentials of the primary Hunting Group port. For example: Port 1, 3 and 5 are members of the same Hunting Group. Port 1 is registered with a SIP account. Ports 3 and 5 are not registered. Ports 3 and 5 will be able to place outbound calls using the SIP account of port 1. Select appropriate value for Hunting Group feature. The original SIP account should be set to Active while the group members should be set to the port number of the Active Port. Example configuration of a multiple Hunting group: FXS Port #1: SIP UserID and Authenticate ID entered, Hunting group set to "**Active**" FXS Port #2: SIP UserID and Authenticate ID left blank, Hunting Group set to "**1**" FXS Port #3: SIP UserID and Authenticate ID left blank, Hunting Group set to "**1**" FXS Port #4: SIP UserID and Authenticate ID entered, Hunting group set to "**Active**" FXS Port #5: SIP UserID and Authenticate ID left blank, Hunting Group set to "**4**" FXS Port #6: SIP UserID and Authenticate ID left blank, Hunting Group set to "**4**" FXS Port #7: SIP UserID and Authenticate ID entered, Hunting group set to "**Active**" FXS Port #8: SIP UserID and Authenticate ID left blank, Hunting Group set to "**7**" Hunting Group 1 contains ports 1, 2, 3. Hunting Group 4 contains ports 4, 5, 6. Hunting Group 7 contains ports 7, 8.
Request URI Routing ID	If configured, device will route the incoming call to designated port by request URI user ID in SIP INVITE.
Enable Port	If set to No, FXS port will become inactive (Default is set to Yes)
Port#	FXS Port Number

Offhook Auto-dial	This feature allows you to automatically dial the number specified in this field as soon as the port is offhooked, i.e. when the receiver on the phone connected to **Port#** is picked up.
Offhook Auto-Dial Delay	Configure the delay time for offhook auto-dial function. Range is 0-60 seconds, default is 0.
Map to FXO Port#	This is used only when peering with a Grandstream GXW410x. Default is 1, Supported values are 1-8, meaning line 1 to line 8 of the GXW410x device where the port will be mapped to.
Map to FXO Gateway IP	This is used when peering with an FXO gateway of any brand. You have to specifically mention the IP and sip port where the call will be sent to.
and Port	Sip port that will be annexed to the IP address above.

TABLE 11: PROFILE SETTINGS

Profile Active	When set to Yes the SIP Profile is activated.
Primary SIP Server	Primary SIP Server's IP address or Domain name provided by VoIP service provider.
Failover SIP Server	Failover SIP Server's IP address or Domain name provided by VoIP Service provider. This server will be used if the Primary SIP server becomes unavailable.
Prefer Primary SIP Server	Default is no. If set to yes it will register to Primary Server if registration with Failover server expires
Outbound Proxy	IP address or Domain name of Outbound Proxy, or Media Gateway, or Session Border Controller. Used by GXW400X for firewall or NAT penetration in different network environments. If symmetric NAT is detected, STUN will not work and ONLY outbound proxy can correct the problem.
SIP transport	User can select UDP or TCP or TLS. Please make sure you're SIP Server or network environment supports SIP over the selected transport method. Default is UDP.

NAT Traversal	This parameter defines whether the GXW400X NAT traversal mechanism is activated or not. If activated (by choosing "Yes") and a STUN server is also specified, then the GXW400X performs according to the STUN client specification. Under this mode, the embedded STUN client will detect if and what type of firewall/NAT is being used. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the GXW400X will use its mapped public IP address and port in all of its SIP and SDP messages. If the NAT Traversal field is set to "Yes" *with no specified STUN server*, the GXW400X will periodically (every 20 seconds) send a blank UDP packet (with no payload data) to the SIP server to keep the "hole" on the NAT open.
DNS Mode	One from the 3 modes are available for "DNS Mode" configuration: -A Record (for resolving IP Address of target according to domain name) -SRV (DNS SRV resource records indicates how to find services for various protocols) -NAPTR/SRV (Naming Authority Pointer according to RFC 2915) -Use Configured IP (Use the three configured IP address instead of any DNS query) One mode can be chosen for the client to look up server. The default value is "A Record"
Primary IP	Configure the primary IP for DNS Mode: Use Configured IP
Backup IP1	Configure the first backup IP for DNS Mode: Use Configured IP
Backup IP2	Configure the second backup IP for DNS Mode: Use Configured IP
Tel URI	The default setting is "Disabled". If the phone has an assigned PSTN Number, this field should be set to "User=Phone" then a "User=Phone" parameter will be attached to the "From header" in the SIP request to indicate the E.164 number. If server supports TEL URI format, then this option needs to be selected.
Use Request Routing ID in SIP INVITE Header	If set to **Yes**, device will use **Request URI Routing ID** defined in FXS ports settings to replace From and Contact headers for outgoing calls.
SIP Registration	This parameter controls whether the GXW400X needs to send REGISTER messages to the proxy server. The default setting is "Yes".
Unregister on Reboot	Default is **No**. If set to "Yes", the SIP user's registration information is cleared on reboot.

Outgoing Call without Registration	Default is **No**. If set to "Yes," user can place outgoing calls even when not registered (if allowed by Internet Telephone Service Provider) but is unable to receive incoming calls. Any port, member of a Hunting Group that is not registered with a SIP account, will be able to place outbound calls using the SIP credentials of the primary Hunting Group port. For example: Port 1, 3 and 5 are members of the same Hunting Group. Port 1 is registered with a SIP account. Ports 3 and 5 are not registered. Ports 3 and 5 will be able to place outbound calls using the SIP account of port 1, even if ***Outgoing Call without Registration*** is set to No
Register Expiration	Allows the user to specify the time frequency (in minutes) for the GXW400X to refresh its registration with the specified registrar. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
Reregister before Expiration	This parameter allows the user to specify the reregistration time before expiration.
Local SIP port	Defines the local SIP port the GXW400X will listen and transmit. The default value for Profile 1 is 5060 and 6060 for Profile 2.
Local RTP Port	Defines the local RTP port pair the GXW400X will listen and transmit. It is the base RTP port for channel 0. When configured, channel 0 will use this port _value for; channel 1 will use port_value+2 for RTP. The default value for Profile 1 is 5004 and 6004 for Profile 2.
Use random port	Default is **No**. If set to Yes, the device will pick randomly-generated SIP and RTP ports. This is usually necessary when multiple GXW400X/HT50X are behind the same NAT.
Hold Target Before Refer	Default is **Yes**. Allows user to hold the phone call before refer it. If set to No, the call will not be hold before referred.
Refer to Use Target Contact	Default is **No**. If set to YES, then for Attended Transfer, the "Refer-To" header uses the transferred target's Contact header information.
Transfer on Conference Hang up	Default is **No**. In which case if conference originator hangs up the conference will be terminated. When option YES is chosen, originator will transfer other parties to each other so that B and C can choose either to continue the conversation or hang up.
Disable Bellcore Style 3-Way Conference	Default is **No**. you can make a Conference by pressing 'Flash' key. If set to **Yes**, you need to dial *23 + second callee number.
Remove OBP from Route Header:	Default is **No**. If set to Yes, the Outbound Proxy will be removed from the route header.

Support SIP Instance ID	Default is Yes. If set to Yes, the contact header in REGISTER request will contain SIP Instance ID as defined in IETF SIP Outbound draft.
Validate incoming SIP message	Default is **No**. If set to yes all incoming SIP messages will be strictly validated according to RFC rules. If message does not pass validation process, call will be rejected.
Check SIP user ID for incoming INVITE	Default is No. Check the SIP User ID in Request URI. If they don't match, the call will be rejected.
Authenticate incoming INVITE	Default is **No.** If set to Yes, device will challenge the incoming INVITE for the Authenticate ID and Password with 401 Unauthorized.
Allow Incoming SIP Messages from SIP Proxy Only	Default is No. If incoming SIP message does not match with SIP Server, it will be rejected.
Use Privacy Header	If set to **Default**, it will only add Privacy or PPI header when special feature is not **Telkom SA** or **CBCOM.**
Use P-Preferred-Identity Header	If set to **Default**, it will only add Privacy or PPI header when special feature is not **Telkom SA** or **CBCOM.**
SIP T1 Timeout	T1 is an estimate of the round-trip time between the client and server transactions. If the network latency is high, select larger value for more reliable usage.
SIP T2 Interval	Maximum retransmission interval for non-INVITE requests and INVITE responses.
SIP Timer D	Set the SIP Timer D. Default is **0**.
DTMF Payload Type	Sets the payload type for DTMF using RFC2833.
Preferred DTMF method (in listed order)	The GXW400X supports up to 3 different DTMF methods including in-audio, via RTP (RFC2833) and via Sip Info. The user can configure DTMF method in a priority list.
Disable DTMF Negotiation	Default is **No**. If set to yes, use above DTMF order without negotiation
Send Hook Flash Event	Default is **No**. If set to yes, flash will be sent as a DTMF event.
Enable Call Features	Default is **Yes**. (If Yes, call features using star codes will be supported locally)
Proxy Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Use NAT IP	NAT IP address used in SIP/SDP message. Default is blank.

Use SIP User-Agent Header	Used to replace SIP User-Agent Header (No Default)
Distinctive Ringtone	Custom Ring Tone 1 to 3 with associate Caller ID: when selected, if Caller ID is configured, then the device will ONLY uses this ring tone when the incoming call is from the Caller ID. System Ring Tone is used for all other calls. When selected but no Caller ID is configured, the selected ring tone will be used for **all** incoming calls. Distinctive ring tones can be configured not only for matching whole number, but also for matching prefixes. In this case symbol * (star) will be used. If server supports Alert-Info header and standard ring tone set (Bellcore) or distinctive ring tone 1-10 is specified, then the ring tone in the Alert-Info header from server will be used. **For example:** If configured as *617, Ring Tone 1 will be used in case of call arrived from Massachusetts. Any other incoming call will ring using cadence defined in parameter System Ring Cadence located under Advanced Settings Configuration page.
Disable Call Waiting	Default is **No**. If set to **YES** Call Waiting indication information will not be provided to analog phone connected to this FXS port.
Disable Call Waiting Caller ID	Default is No. If set to YES Call Waiting caller ID will not be provided to analog phone connected to this FXS port.
Disable Call Waiting Tone	Default is **No**. This is to disable the stutter Call Waiting Tone when a Call Waiting call arrives. The CWCID will still be displayed.
Disable Reminder Ring for On-Hold Call	Default is No. This is to disable the Reminder Ring that is played when a call is waiting on hold and the analog phone goes on-hook.
Disable Visual MWI	Visual message indicator is a special on-hook caller ID type message that enables and disables the message waiting light on certain phones. GXW400X has this feature enabled by default. However, certain phones (rare) that do not support it may mistakenly treat this CID signal as an incoming call. A configuration option is needed to turn on MWI in this case.
Do Not Escape '#' as %23 in SIP URI	If set to "**Yes**", device will use '#' instead of %23 in the send URI.
Disable Multiple m line in SDP	Default is **No**. If set to Yes, device will send only one m line in SDP, regardless how many m field in the incoming SDP.
Ring Timeout	Incoming call will stop ringing when not picked up given a specific period of time.

Hunting Group Ring Timeout	Default is **20 seconds**. If call is not answered within this designated time period, the call will be forwarded to the next member of a Hunt Group.
Hunting Group Type	Linear and Circular. Linear style will sort the call to the lowest-numbered available line, this is also called “serial hunting”. Circular style will distribute the calls "round-robin". If a call is assigned to line 1, the next call goes to 2 and the next to 3. The succession throughout each of the lines continues even if one of the previous lines becomes available. When the end of the hunt group is reached, the hunting starts over at the first line. Lines are skipped if they are still busy on a previous call. These two hunting styles can be configured from the Profile_x page.
Delayed Call Forward Wait Timeout	Default value is **20 seconds**. In case this feature activated using * codes (*92 code), the call will be forwarded after this preconfigured amount of time.
No Key Entry Timeout	Default is **4 seconds**. Call will be completed within this time interval if no additional key entry occurs.
Early Dial	Default is **No**. Use only if proxy supports 484 response. This parameter controls whether the phone will send an early INVITE each time a key is pressed when a user dials a number. If set to “Yes”, an INVITE is sent using the dial-number collected thus far; Otherwise, no INVITE is sent until the “(Re-)Dial” button is pressed or after about 5 seconds have elapsed if the user forgets to press the “Re-Dial” button. The “Yes” option should be used ONLY if there is a SIP proxy configured and the proxy server supports 484 Incomplete Address response. Otherwise, the call will likely be rejected by the proxy (with a 404 Not Found error). *This feature is NOT designed to work with and should NOT be enabled for direct IP-to-IP calling*.
Dial Plan Prefix	Sets the prefix added to each dialed number.
Use # as Dial Key	Allows users to configure the “#” key as the “Send” (or “Dial”) key. If set to “Yes”, “#” will send the number. In this case, this key is essentially equivalent to the “Dial” key. If set to “No”, this “#” key can be included as part of number.

Dial Plan	**Dial Plan Rules**: 1. Accept Digits: 1,2,3,4,5,6,7,8,9,0 , *, #, A,a,B,b,C,c,D,d 2. Grammar: x - any digit from 0-9; a. xx+ - at least 2 digits number; b. xx. – at least 2 digits number; c. ^ - exclude; d. [3-5] - any digit of 3, 4, or 5; e. [147] - any digit 1, 4, or 7; f. <2=011> - replace digit 2 with 011 when dialing g. < =1> - add a leading 1 to all numbers dialed, vice versa will remove a 1 from the number dialed h. | - or • **Example 1:** {[369]11 | 1617xxxxxxx} – Allow 311, 611, 911, and any 10 digit numbers of leading digits 1617 • **Example 2:** {^1900x+ | <=1617>xxxxxxx} – Block any number of leading digits 1900 and add prefix 1617 for any dialed 7 digit numbers • **Example 3:** {1xxx[2-9]xxxxxx | <2=011>x+} – Allow any length of number with leading digit 2 and 10 digit-numbers of leading digit 1 and leading exchange number between 2 and 9; If leading digit is 2, replace leading digit 2 with 011 before dialing. 3. Default: Outgoing - {x+} Example of a simple dial plan used in a Home/Office in the US: **{ ^1900x. | <=1617>[2-9]xxxxxx | 1[2-9]xx[2-9]xxxxxx | 011[2-9]x. | [3469]11 }** ***Explanation*** of example rule (reading from left to right): • **^1900x. -** prevents dialing any number started with 1900 • **<=1617>[2-9]xxxxxx -** allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically • **1[2-9]xx[2-9]xxxxxx |-** allows dialing to any US/Canada Number with 11 digits length • **011[2-9]x. -** allows international calls starting with 011 **[3469]11 -** allow dialing special and emergency numbers 311, 411, 611 and 911 **Note:** In some cases user wishes to dial strings such as *123 to activate voice mail or other application provided by service provider. In this case * should be predefined inside dial plan feature and the Dial Plan will be: **{ [x*]+ }**.

Subscribe for MWI	Default is **No**. When set to "Yes" a SUBSCRIBE for Message Waiting Indication will be sent periodically.
Send Anonymous	If this parameter is set to "Yes", the "From" header along with Privacy and P_Asserted_Identity headers in outgoing INVITE message will be set to anonymous, blocking Caller ID.
Anonymous Call Rejection	Default is **No**. If set to Yes, incoming calls with anonymous Caller ID will be rejected with 486 Busy message.
Special Feature	Default is **Standard**. Choose the selection to meet some special requirements from Softswitch vendors. Example of vendors - CBCOM, RNK.
Session Expiration	Grandstream implemented SIP Session Timer. The session timer extension enables SIP sessions to be periodically "refreshed" via a SIP request (UPDATE, or re-INVITE. When the session interval expires, if there is no refresh via a UPDATE or re-INVITE message, the session will be terminated. Session Expiration is the time (in seconds) at which the session is considered timed out, if no successful session refresh transaction occurs beforehand. The default value is **180 seconds**.
Min-SE	The minimum session expiration (in seconds). The default value is **90 seconds**.
Caller Request Timer	If selecting "Yes" the phone will use session timer when it makes outbound calls if remote party supports session timer.
Callee Request Timer	If selecting "Yes" the phone will use session timer when it receives inbound calls with session timer request.
Force Timer	If selecting "Yes" the phone will use session timer even if the remote party does not support this feature. Selecting "No" will allow the phone to enable session timer only when the remote party support this feature. To turn off Session Timer, select "No" for Caller Request Timer, Callee Request Timer, and Force Timer.
UAC Specify Refresher	As a Caller, select UAC to use the phone as the refresher, or UAS to use the Callee or proxy server as the refresher.
UAS Specify Refresher	As a Callee, select UAC to use caller or proxy server as the refresher, or UAS to use the phone as the refresher.
Force INVITE	Session Timer can be refreshed using INVITE method or UPDATE method. Select "Yes" to use INVITE method to refresh the session timer.

Send Re-INVITE After Fax	Default is **No**, If set to "Yes", device will send an INVITE with audio vocoders upon completition of Fax to continue session in audio only.
Enable Silence Detection for Fax Disconnect	For fax machines that do not send a Disconnect when fax is done. This option Enables/Disables the detection of silence in order to know the fax has finished. The silence period is non-configurable and fixed to 7 seconds.
Enable 100rel	Enables the use of PRACK (Provisional Acknowledgment) method.
Add Auth Header On Initial REGISTER	If set to **Yes**, device will include authorization header in the Register request.
Use First Matching Vocoder in 200OK SDP	Default is **No**. If set to "Yes", device will include only the first match vocoder in its 200OK response, otherwise it will include all match vocoders in same order received in INVITE.
Preferred Vocoder	The GXW400X supports up to 5 different Vocoder types including G.711 A-/U-law, G.726 (Supports bit rates 16, 24, 32 and 40), G.723.1, G.729A/B/E and iLBC. The user can configure Vocoders in a preference list that will be included with the same preference order in SDP message. The first Vocoder is entered by choosing the appropriate option in "Choice 1". The last Vocoder is entered by choosing the appropriate option in "Choice 8".
Voice Frames per TX	Default is 2, from 1 -4 for G711/G726/G729
G723 Rate	Defines the encoding rate for G.723 vocoder. By default, 6.3kbps rate is chosen.
iLBC Frame Size	Sets the iLBC frame size in 20ms or 30ms
iLBC Payload type	Default value is **97**. Defines payload type for iLBC. The valid range is between 96 and 127.
AAL2-G726-16 Payload type	Default value is **100**. Range is from 96 to 127.
AAL2-G726-24 Payload type	Default value is **99**. Range is from 96 to 127.
AAL2-G726-32 Payload type	Default value is **104**. Range is from 96 to 127.
AAL2-G726-40 Payload type	Default value is **103**. Range is from 96 to 127.
G729E payload type	Default value is **102**. Range is from 96 to 127.

VAD	Default is **No**. VAD allows detecting the absence of audio and conserve bandwidth by preventing the transmission of "silent packets" over the network.
Symmetric RTP	Default is **No**. When set to Yes the device will change the destination to send RTP packets to the source IP address and port of the inbound RTP packet last received by the device.
Fax Mode	T.38 (Auto Detect) FoIP by default, or Pass-Through (must use codec PCMU/PCMA)
Fax Tone Detection Mode	Default is Callee. This decides whether Caller or Callee sends out the re-INVITE for T.38 or Fax Pass Through.
Jitter Buffer Type	Select either Fixed or Adaptive based on network conditions.
Jitter Buffer Length	Select Low, Medium or High based on network conditions. • High (initial 200ms, min 40ms, max 600ms) Note: not all vocoders can meet the high requirement • Medium (initial 100ms, min 20ms, max 200ms) • Low (initial 50ms, min 10ms, max 100ms)
SRTP Mode	Default is **Disabled**. Other options are **Enabled but not forced**, and **Enabled and forced**. It uses SDP Security Description to exchange key. Please refer SDES: http://www.apps.ietf.org/rfc/rfc4568.html SRTP: http://www.apps.ietf.org/rfc/rfc3711.html
Crypto Life Time	Default is **Enabled**. Allows user to enable or disable Crypto life time when using SRTP.
SLIC Setting	Depends on standard phone type (and location).
Caller ID Scheme	Select the value according to the local Telco standard where the GXW400X is installed. Please refer to the pull down list to select.
Polarity Reversal	Default is **No**. If set to “Yes”, polarity will be reversed upon call establishment and termination.
Loop Current Disconnect	Set to Yes if the traditional PBX you are using with GXW400X uses this method for signaling call termination. Default is **No**.
Play busy/reorder tone before Loop Current Disconnect	Default is **No**. If set to Yes, it will play busy/reorder tone before loop current disconnect upon call fail.

Loop Current Disconnect duration	Default is **200**. In 100 - 1 0000 milliseconds range.
Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Enable Hook Flash	Default is **Yes**. If set to "No", FLASH button could only be used for terminating calls.
Hook Flash timing	Time period when the cradle is pressed (Hook Flash) to simulate FLASH. To prevent unwanted activation of the Flash/Hold and automatic phone ring-back, adjust this time value.
On Hook Timing	On-hook timing is the minimum time for an on-hook event to be validated.
Gain	Handset volume adjustment. RX is for receiving volume (direction FXS→to analog phone), TX is for transmission volume (Analog phone→ to FXS). Default values are **0dB** for both parameters. Loudest volume: +6dB Lowest volume: -6dB.
Ring Tones	Configure ring cadences according to preference.

SAVING THE CONFIGURATION CHANGES

After user makes a change to the configuration, press the "Update" button in the Configuration Menu. The web browser will then display a message window to confirm saved changes.

Grandstream recommends reboot or power cycle the IP phone after saving changes.

REBOOTING FROM REMOTE

Press the "Reboot" button at the bottom of the configuration menu to reboot the phone remotely. The web browser will then display a message window to confirm that reboot is underway. Wait 30 seconds to log in again.

CONFIGURATION THROUGH A CENTRAL SERVER

Grandstream GXW400X can be automatically configured from a central provisioning system.

When GXW400X boots up, it will send TFTP or HTTP/HTTPS requests to download configuration files, "cfg000b82xxxxxx" and "cfg00082xxxxxx.xml", where "000b82xxxxxx" is the LAN MAC address of the GXW400X. If the download of "cfgxxxxxxxxxxxx.xml" is not successful, the provision program will issue

request a generic configuration file “cfg.xml”. Configuration file name should be in lower case letters.

The configuration data can be downloaded via TFTP or HTTP/HTTPS from the central server. A service provider or an enterprise with large deployment of GXW400X can easily manage the configuration and service provisioning of individual devices remotely from a central server.

Grandstream provides a central provisioning system GAPS (Grandstream Automated Provisioning System) to support automated configuration of Grandstream devices. GAPS uses enhanced (NAT friendly) TFTP or HTTP (thus no NAT issues) and other communication protocols to communicate with each individual Grandstream device for firmware upgrade, remote reboot, etc.

Grandstream provides GAPS service to VoIP service providers. Use GAPS for either simple redirection or with certain special provisioning settings. At boot-up, Grandstream devices by default point to Grandstream provisioning server GAPS, based on the unique MAC address of each device, GAPS provision the devices with redirection settings so that they will be redirected to customer’s TFTP or HTTP/HTTPS server for further provisioning.

Grandstream also provides configuration tools (Windows and Linux/Unix version) to facilitate the task of generating device configuration files. The Grandstream configuration tools are free to end users. The configuration tools and configuration templates are available for download from http://www.grandstream.com/support/tools .

SOFTWARE UPGRADE

Software upgrade can be done via either TFTP or HTTP/HTTPS. The corresponding configuration settings are in the ADVANCED SETTINGS configuration page.

FIRMWARE UPGRADE THROUGH TFTP/HTTP/HTTPS

To upgrade via TFTP or HTTP/HTTPS, the "Firmware Upgrade and Provisioning upgrade via" field needs to be set to TFTP HTTP or HTTPS, respectively. "Firmware Server Path" needs to be set to a valid URL of a TFTP or HTTP server, server name can be in either FQDN or IP address format. Here are examples of some valid URL.

e.g. firmware.mycompany.com:6688/Grandstream/1.0.9.1

e.g. firmware.grandstream.com

NOTES:

- Firmware upgrade server in IP address format can be configured via IVR. Please refer to the CONFIGURATION GUIDE section for instructions. If the server is in FQDN format, it must be set via the web configuration interface.

- Grandstream recommends end-user use the Grandstream HTTP server. Its address can be found at http://www.grandstream.com/support/firmware . Currently the HTTP firmware server URL is firmware.grandstream.com. For large companies, we recommend to maintain their own TFTP/ HTTP/HTTPS server for upgrade and provisioning procedures.

- Once a "Firmware Server Path" is set, user needs to update the settings and reboot the device. If the configured firmware server is found and a new code image is available, the GXW will attempt to retrieve the new image files by downloading them into the GXW400x 's SRAM. During this stage, the GXW's LEDs will blink until the checking/downloading process is completed. Upon verification of checksum, the new code image will then be saved into the Flash. If TFTP/HTTP/HTTPS fails for any reason (e.g. TFTP/HTTP/HTTPS server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the GXW will stop the TFTP/HTTP/HTTPS process and simply boot using the existing code image in the flash.

- Firmware upgrade may take as long as 15 to 30 minutes over Internet, or just 5 minutes if it is performed on a LAN. It is recommended to conduct firmware upgrade in a controlled LAN environment if possible. For users who do not have a local firmware upgrade server, Grandstream provides a NAT-friendly TFTP server on the public Internet for firmware upgrade.

- Grandstream's latest firmware is available http://www.grandstream.com/support/firmware .

  Oversea users are strongly recommended to download the binary files and upgrade firmware locally in a controlled LAN environment.

- Alternatively, user can download a free TFTP or HTTP server and conduct local firmware upgrade. A free windows version TFTP server is available for download from http://support.solarwinds.net/updates/New-customerFree.cfm. Our latest official release can be downloaded from http://www.grandstream.com/y-firmware.htm.

INSTRUCTIONS FOR LOCAL FIRMWARE UPGRADE:

1. Unzip the file and put all of them under the root directory of the TFTP server.
2. Put the PC running the TFTP server and the GXW400X device in the same LAN segment.
3. TFTP server's security settings should be changed from "Receive Only" to "Transmit Only" for the firmware upgrade.
4. Configure the Firmware Server Path with the IP address of the PC
5. Update the change and reboot the unit.

CONFIGURATION FILE DOWNLOAD

Grandstream SIP Device can be configured via Web Interface as well as via Configuration File through TFTP or HTTP/HTTPS. "Config Server Path" is the TFTP or HTTP/HTTPS server path for configuration file. It needs to be set to a valid URL, either in FQDN or IP address format. The "Config Server Path" can be same or different from the "Firmware Server Path".

A configuration parameter is associated with each particular field in the web configuration page. A parameter consists of a Capital letter P and 2 to 3 (Could be extended to 4 in the future) digit numeric numbers. i.e., P2 is associated with "Admin Password" in the ADVANCED SETTINGS page. For a detailed parameter list, please refer to the corresponding firmware release configuration template.

When a Grandstream device boots up or reboots, it will issue a request for a configuration file "cfgxxxxxxxxxxxx", where "xxxxxxxxxxxx" is the MAC address of the device, i.e., "cfg000b820102ab". In addition, device will also requests a XML configuration file "cfgxxxxxxxxxxxx.xml". If the download of "cfgxxxxxxxxxxxx.xml" is not successful, the provision program will issue a request for a generic configuration file "cfg.xml". Configuration file name should be in lower case letters.

For more details on Grandstream Device provisioning, please refer to:
http://www.grandstream.com/general/gs_provisioning_guide_public.pdf

FIRMWARE AND CONFIGURATION FILE PREFIX AND POSTFIX

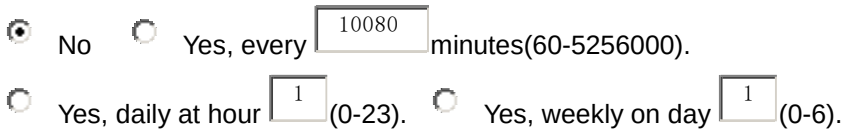
Firmware Prefix and Postfix allows device to download the firmware name with the matching Prefix and Postfix. This makes it possible to store ALL of the firmwares with different version in one single directory. Similarly, Config File Prefix and Postfix allows device to download the configuration file with the matching Prefix and Postfix. Thus multiple configuration files for the same device can be stored in one directory.

In addition, when the field "Check New Firmware only when F/W pre/suffix changes" is selected, the device will only issue firmware upgrade request if there are changes in the firmware Prefix or Postfix.

MANAGING FIRMWARE AND CONFIGURATION FILE DOWNLOAD

When "Automatic Upgrade" is set "**Yes, every**" the auto check will be done in the minute specified in this field. If set to "**daily at hour (0-23)"**, Service Provider can use P193 (Auto Check Interval) to have the devices do a daily check at the hour set in this field with either Firmware Server or Config Server. If set to "**weekly on day (0-6)"** the auto check will be done in the day specified in this field. This allows the device periodically check if there are any new changes need to be taken on a scheduled time. By defining different intervals in P193 for different devices, Server Provider can spread the Firmware or Configuration File download in minutes to reduce the Firmware or Provisioning Server load at any given time.

Automatic Upgrade:



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RESTORE FACTORY DEFAULT SETTING

WARNING! Restoring the Factory Default Setting will DELETE all configuration information of the phone. Please BACKUP or PRINT out all the settings before you approach to following steps. Grandstream will not take any responsibility if you lose all the parameters of setting and cannot connect to your VoIP service provider.

FACTORY RESET

There are two (2) methods for resetting your unit:

RESET BUTTON

Reset default factory settings following these four (4) steps:

1. Unplug the Ethernet cable.
2. Locate a needle-sized hole on the back panel of the gateway unit next to the power connection.
3. Insert a pin in this hole, and press for about 7 seconds.
4. Take out the pin. All unit settings are restored to factory settings.

IVR COMMAND

Reset default factory settings using the IVR Prompt (Table 5):

1. Dial "***" for voice prompt.
2. Enter "99" and wait for "reset" voice prompt.
3. Enter the encoded MAC address (Look below on how to encode MAC address).
4. Wait 15 seconds and device will automatically reboot and restore factory settings.

Encode the MAC Address

1. Locate the MAC address of the device. It is the 12 digit HEX number on the bottom of the unit.
2. Key in the MAC address. Use the following mapping:

0-9:	0-9
A:	22 (press the “2” key twice, “A” will show on the LCD)
B:	222
C:	2222
D:	33 (press the “3” key twice, “D” will show on the LCD)
E:	333
F:	3333

For example: if the MAC address is 000***b***8200***e***395, it should be keyed in as “000***222***8200***333***395”.

NOTE:

1. Factory Reset will be disabled if the “**Lock keypad update**” is set to “Yes”.
2. Please be aware by default the GXW400X WAN side HTTP access is disabled. After a factory reset, the device’s web configuration page can be accessed only from its LAN port.